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TITLE: Speech and data multiplexor optimized for use over impaired and bandwidth restricted analog channels

Abstract Text (1):
The present invention relates to an apparatus and method for the simultaneous transmission of analog speech and modulated data, such apparatus and method being optimized for use over impaired and bandwidth restricted analog channels, or digital representations of such channels. In each instance of use, an evaluation is made of the available channel bandwidth, with a frequency division multiplex scheme allocating a voice sub-band, with data transmission allocated to sub-bands above, below, or around, this selected voice sub-band. The speech and data sub-band allocations are made by the multiplexor in response to user input of either a requested speech quality, a requested data rate, or a value indicating the relative user weighting of speech quality and data rate. A multi-carrier multi-mode modulation scheme is employed for data transmission, with this scheme having the ability to fully utilize the remaining bandwidth, and further, being capable of adapting to the impairments most likely present on the fringes of bandwidth restricted analog channels. When the analog channel employed is the standard voice-grade telephone circuit, good speech quality simultaneously with 3000 bps data transmission may be expected. The further ability to automatically switch to full bandwidth data transmission when voice transmission is not being attempted is also provided.

Brief Summary Text (7):
The class of methods within which the present invention falls generally is the class which relates to the division in frequency of the available bandwidth, with speech and data signals assigned to separate frequency sub-bands.

Brief Summary Text (9):
Another approach in the prior art is to employ the frequencies between 0 and 200 Hz and a single carrier modulation scheme to produce a 200-300 bps data transmission rate. A disadvantage of this method is that single carrier transmission is sensitive to the composite noise over the entire sub-band employed, and in this case power equipment induced 60 Hz and 180 Hz noise (Bell System Technical Journal, Vol. 63, No. 5, pp. 775-818) will cause signal distortion. This distortion limits the information carrying ability of the carrier and results in a relatively low data rate. A further disadvantage is that frequencies below 200 Hz are either unavailable or severely attenuated on long distance public switched telephone network links (BSTJ Vol. 63, No. 5, pp. 775-818).

Brief Summary Text (10):
A similar approach in the prior art is disclosed in U.S. Pat. No 4,011,407 issued to N. DiSanti and F. Oster on Feb. 26, 1976, in which a single data carrier is employed in a 100 Hz band centered at 2900 Hz. The data rate achieved in this case is 250 bps with indications that 500 bps is possible with a doubling of the data bandwidth to 200 Hz. This low data rate is a consequence of the inability to utilize a broader sub-band with a single carrier modulation scheme, since such a broader band would be subject to frequency dependent attenuation and group delay (BSTJ Vol. 63, No. 9, pp. 2059-2119), and single carrier modulation is sensitive to such impairments.

Brief Summary Text (11):
A third approach in the prior art is that disclosed in U.S. Pat. No. 4,546,212, issued to J. Crowder, Sr., on Oct. 8, 1985, for use over the public telephone network in which two sub-bands are proposed, 0-800 Hz for speech, and 800 Hz and above for data. The disadvantages of this scheme are first, that the speech quality is very poor due to the severely limited bandwidth allowed for voice transmission, and second, that a switching means must be employed to allow full bandwidth access in order that DTMF (dual tone multi-frequency) call signalling tones (which spans the 800 Hz limits) will be possible.

Brief Summary Text (15):
It is a further object of the invention to provide a means for successful transmission of data, in one or both of two data sub-bands, where one sub-band borders the upper available channel band edge and the other borders the lower available channel band edge, in which sub-bands the common impairments of induced noise, attenuation, and group delay, may be present.

Brief Summary Text (16):
Another object of the present invention is to provide a means for each user to obtain an optimized combination of speech quality and data throughput, in accordance with that user's preferences At one limit, in which voice quality is the dominant user preference, an object of the present invention is the provision of a mechanism for preselection of the voice sub-band. At the other limit, in which data rate is the overriding user preference, an object of the present invention is the provision of a mechanism for user selection of the data rate. In this situation the speech and data multiplexor will determine the width of the data sub-band or sub-bands required to sustain the requested data rate. Inside these two limits, it is an object of the present invention to provide a mechanism for notification of the speech and data multiplexor of the user's relative weighting of voice quality and data rate, and to provide a mechanism through which the speech and data multiplexor will respond to this information in the allocation of the voice sub-band.

Brief Summary Text (19):
It is yet another object of the present invention to provide embodiments in which the bandwidth restricted analog channel may be, in one case, a loaded (or unloaded) local telephone loop

between a subscriber station and the telephone central office, and in a separate case, a general public switched telephone network circuit. In the latter case it is an object of the present invention to include as the bandwidth restricted analog channel, the digital representations of such channels, for example by pulse code modulation (PCM) or adaptive differential pulse code modulation (ADPCM), that are present within the public switched telephone network.

Brief Summary Text (23):
In the embodiment in which the analog channel is a public switched telephone network circuit, both local and remote units are identical and are denoted as subscriber units.

Brief Summary Text (24):
In the embodiment in which the analog channel is a loaded (or unloaded) local loop, the remote unit is located in the telephone company central office and is denoted as a central office unit. In this case the data port is connected to a telephone company data service unit, and the speech port is connected to the telephone company voice switch, and there are minor signal sensing and replication differences between subscriber and central office units that will be discussed in detail below.

Brief Summary Text (26):
The first component is a multi-carrier, multi-mode, ensemble modem as disclosed in U.S. patent application Ser. No. 06-736,200 filed May 20, 1985 entitled "Ensemble Modem Structure for Imperfect Transmission Media" and which is assigned to the same assignee as the assignee of the present invention. The application serial number 06-736,200 is incorporated by reference in this application in accordance with the provisions of section 608.01(p) of the Manual of Patent Examining Procedure of the U.S. Patent and Trademark Office. As described in detail in application Ser. No. 06-736,200, this component provides the ability to separately determine the response of the analog channel to the transmission of 512 possible signal carriers, uniformly spaced 7.8125 Hz apart, across the frequency range from zero to 4000 Hz. This ability provides the mechanism for the determination by the speech and data multiplexors of the available analog channel bandwidth. Further, the ensemble modem has the ability to optimally allocate its allowable-transmission power in such a way as to choose carriers, and the modulation scheme employed on each, so that the overall data rate achieved is maximized. This selection procedure is based on the background noise and transmission loss experienced by each carrier and hence will adapt to induced power equipment noise in the low frequency sub-band, if such a band was selected for transmission. The carrier by carrier demodulation of the ensemble modem permits additional compensation for frequency dependent attenuation in both low and high frequency sub-bands. Finally the long symbol period modulation with its associated guardtime makes the transmission insensitive to frequency dependent group delay. Together these characteristics of the ensemble modem provide the superior performance, with respect to data transmission, of the present invention.

Brief Summary Text (27):

The second basic component of the speech and data multiplexor is comprised of the speech and channel port interfaces and the voice and data band pass filters. The speech and channel port interfaces are standard hybrid circuits as employed in the telephone industry. The filter stages determine the location of the lower edge and the upper edge of the voice sub-band. (In embodiments in which only one data sub-band is present there is only one voice frequency limit to be determined.) Their purpose is, upon transmission and reception, to separate the voice and data energy, to insure no data transmission interference with speech reception and vice-versa. In a selected embodiment these filter edges may be controlled by the user through the data port, providing the ability to select the sub-band for speech transmission.

Brief Summary Text (28):
The final basic component of the speech and data multiplexor is comprised of those circuit components required to ensure that no impairment is made to those signals that would be present in the circumstance in which the analog transmission channel in use is a standard public switched telephone network circuit. These circuit components are designed to detect for a subscriber unit, at the speech port (which in this circumstance is a telephone port), on and off hook states, and to detect at the channel port, ringing voltage and loop battery. They are also designed to then reproduce, at the telephone port, loop battery and ringing voltage, and, at the channel port, the on and off hook states. (For the embodiment in which a central office unit is present, the ports at which all detection and reproduction functions take place are reversed). This last basic functional component also provides the ability, through the offhook detection circuit, to determine if a voice call is not in progress, and hence the ability to automatically notify the ensemble modem component when the full bandwidth is available for data transmission.

Drawing Description Text (6):
FIG. 4 is a diagram illustrative of the analog channel frequency band division for the preferred embodiment in which data sub-bands arbund the voice sub-band are employed.

Drawing Description Text (7):
FIG. 5 is a diagram illustrative of the analog channel frequency band division for the preferred embodiment in which data sub-band above the voice sub-band is employed.

Drawing Description Text (8):
FIG. 6. is a diagram illustrative of the analog channel frequency band division for the preferred embodiment in which data sub-band below the voice sub-band is employed.

Drawing Description Text (9):
FIG. 7 is a diagram illustrative of the analog channel frequency band division for the preferred embodiment in which a data sub-band above the voice sub-band is employed, and in which automatic switching to full bandwidth data transmission is made in the absence of a speech connection.

Drawing Description Text (11):

FIG. 9 is a block diagram of the simplified architecture of the preferred embodiment in which a fixed data sub-band above the voice sub-band is employed, and in which no switching to full bandwidth data transmission is possible.

Detailed Description Text (17):
The ensemble modem component 49 must be directed, either by instructions stored in program ROM or RAM, or through user input from the data port 3, of the user voice quality and data rate preferences. Based on these instructions or input, the multiplexor determines the voice and data sub-bands in a fashion described in detail below. During data transmission the ensemble modem has the capability to use 512 carriers, spaced 7.8125 Hz apart from zero to 4000 Hz, and those carriers within the selected voice sub-band must now be eliminated from the modulation process. This substantially prevents the ensemble modem from adding energy resident within the voice sub-band to the data transmit path 65. Any energy components in this voice sub-band that are introduced by signal changes during symbol transitions are then removed by the data band pass filters 35.

Detailed Description Text (18):
The ensemble modem utilizes two special carriers, called pilots, whose preferred locations are approximately 1450 Hz and 1700 Hz. In the present invention these pilots must be moved to either the lower or upper data sub-band, and, depending on the available width and transmission quality of these bands, the pilot frequency separation may require reduction.

Detailed Description Text (19):
During the training sequence the ensemble modem measures the background noise present at each carrier frequency and the attenuation suffered by each carrier. Based on this information, the ensemble modem computes the modulation scheme (for example 2, 4, or 6 bits) and power allocated to each available carrier, in accordance with the waterfill algorithm as described in pending application Ser. No. 06-736,200. In the present invention this procedure plays an even more important role than it does in full bandwidth transmission. This is because in the full bandwidth situation the central portion of the available band is usually unimpaired, and power and bits per carrier allocations are quite uniform as a result, whereas the sub-bands employed in the present invention, lying on the fringes of a restricted channel, usually suffer significant impairment. In addition, the full bandwidth situation often sees all available power expended on carriers in the central band, while in the present invention with only fringe bands available for use, the training sequence provides a more detailed determination of the total bandwidth available on a restricted channel.

Detailed Description Text (21):
It is possible to allow the preference instructions contained in either the originating or the answering multiplexor to control the sub-band selections, but for this description we assume control to reside with the originating modem which is further assumed to be the local speech and data multiplexor. At call origination, the orignating modem will determine by inspection of internal data

memory whether the answering modem will respond in accordance with the local user's preference information. When this is the case, the training sequence performed depends on the remaining stored user preference information. In the situation where the user requests a particular voice sub-band, training is performed outside this sub-band in the normal fashion. If the user has requested a particular data rate, a series of training sequences is performed, beginning with a voice sub-band selection obtained from an internal table relating expected data rates and voice sub-bands. Training is repeated with larger or smaller voice sub-bands in pre-determined increments, until a data rate equal to the selected data rate, to within a user specified tolerance, is achieved, or until all the bandwidth available, or in the case that the analog channel is a public switched telephone network channel, all the bandwidth available consistent with the unimpaired transmission of all telephone network signalling has been allocated without the requested data rate having been achieved. In the latter case, the user is requested through the multiplexor data port, to resubmit a data rate request.

Detailed Description Text (22):
The final case is that in which the user has supplied his relative preference for voice quality and data rate through input of a single value within a predetermined range. (For example, an integer between zero and ten with larger integers indicating voice quality preference over data rate.) In this case, training is performed outside a voice sub-band determined by inspection of an internal data table relating preference values and voice sub-bands.

Detailed Description Text (23):
If the remote multiplexor is not known to have the local user's preference information, the two ensemble modems will train over a fixed, narrow sub-band above a fixed, wide voice sub-band and the local user's preference information will be transferred to the remote multiplexor. At this point the two multiplexors will initiate a new training sequence as described above. In all cases, when training is complete, data transmission may begin.

Detailed Description Text (25):
Finally we indicate that in preferred embodiments in which the voice sub-band (and hence the data sub-band edge or edges) is not variable, that a reduction in the processing requirements of th ensemble modem is possible. For example, if a single fixed data sub-band above the voice sub-band were to be used it would be appropriate to alter the sample rates of the analog to digital converter 43 and the digital to analog converter 52 illustrated in FIG. 8. The purpose of this alteration would be upon transmission to generate a fundamental frequency sub-band of width suitable for the data transmission, with frequency aliased copies of this sub-band then also present in the transmit path 65. The data band pass filter stage would be designed to remove all but a selected one of these replicated sub-bands, with the remaining selected sub-band carrying the modulated data for transmission. At the receiving multiplexor, after passage through the data band pass filter stage 37, the signal would be multiplied in the gain stage 53 by a single frequency tone centered Within the data sub-band and the result presented to the analog to digital converter 52, which

would operate at the lowest sample rate sufficient to recover the modulated data carriers. As an example of the reduced processing this approach would allow, we consider a 64 point complex inverse Fast Fourier Transform to be presented to a digital to analog converter that is chosen to operate at a 1600 Hz sample rate. This would produce an 80 millisecond packet with carrier spacing of 12.50 Hz. With the NlogN estimate of FFT processing requirements this would produce a per packet processing reduction of 12 times over a full 512 point FFT, and only 64 carriers would be modulated with data.

Detailed Description Text (28):
The speech energy is filtered through the voice band pass filtering stage 31 to remove all speech energy in those sub-bands that have been pre-selected for data transmission.

Detailed Description Text (29):
The data enters the ensemble modem through data port 3, and this data is modulated by said modem and placed on the data transmit path 65. This signal then passes through the data band filtering stage 35 in which all data energy present in the preselected voice frequency sub-band is removed. Both voice and data signals are now present on the signal path 69 and are converted for transmission over the analog channel by the two-to-four wire interface 29 and are output through the channel port 7.

Detailed Description Text (30):
After transmission through the analog channel 9 the composite voice and data signal is received at the remote channel port 11 of the remote speech and data multiplexor 13 (see FIG. 1). As noted above, the port 11 of the multiplexor 13 corresponds in function to the port 7 of the multiplexor 5 in FIG. 1. Since the components to be described of the remote speech and data multiplexor 13 are common to the local speech and data multiplexor 5, reference will continue to be made to FIG. 8. In this regard the composite data signal is then converted by the two-to-four wire interface 29 and is placed on the receive path 71. This composite signal is then processed by two filter stages. In the first case, the signal is processed by the data band filter stage 37 to remove from the signal all energy present within the voice sub-band frequency; and the resulting signal (containing only frequencies within the data sub-band or sub-bands) is then presented to the data receive path 67 for processing by the ensemble modem 49. The modem demodulates the receive signal and transmits the data to the data port 3.

Detailed Description Text (31):
Similarly, the composite signal is processed by the voice band pass filter stage 33 to remove those frequency components in the signal residing outside the preselected voice sub-band. After such processing, the voice signal is presented to the receive path 57 for conversion by the two-to-four wire speech interface 27 and presentation to the speech port 1.

Detailed Description Text (32):
In accordance with the present invention all filter stages 31, 33, 35 and 37 interface with the ensemble modem 49 and allow the ensemble modem to determine the edge frequencies of the sub-bands

to be employed for voice and data transmission. This interface is accomplished by control lines between the ensemble modem and the various filter stages, and for simplicity, these control lines have not been included in FIG. 8.

Detailed Description Text (42):
In the central office environment the data port 17 presents and receives the transmitted and received data through a data service unit 25 as supplied by the telephone company. The transmission and reception beyond this data service unit 25 will take place in a fashion as determined by the telephone company.

Detailed Description Text (43):
FIG. 9 illustrates an embodiment in which a simplified architecture is employed. In this architecture a subscriber speech and data multiplexor 5 communicates through the speech port 1 and the channel port 7 by means of a passive bidirectional voice filter 59. This filtering stage is designed for that case in which only a single data sub-band above the voice is present. In this instance, a separate passive bidirectional filter stage 61 is employed to separate the data sub-band frequencies from the voice sub-band frequencies. The benefit of this architecture is primarily cost reduction. This occurs as a result of the elimination of all telephone signalling sensing and replication circuits present and illustrated in FIG. 8. This elimination is possible since the passive low pass filter stage 59 is transparent to all these sensing and signalling functions. In this architecture the ensemble modem 63 is precisely the ensemble modem 26 as previously referred to in FIG. 3 of the referenced applicaton Ser. No. 06/736,200. In the simplest implementation of this embodiment the selection of the voice and data sub-band separation frequency would not be programmable by the ensemble modem 49 shown in FIG. 8 of the present applicaiton. However, an alternate implementation would include multiple circuit components within the bi-directional filter stage 59 of FIG. 9, and control lines between this stage and the ensemble modem 63 allowing the selection of filter components, and hence the selection of the voice and data sub-band separation frequency.

Detailed Description Text (44):
FIGS. 4, 5, and 6 indicate schematically the frequency band subdivision employed in preferred embodiments of the present invention. In each case, indication is made of the frequency sub-band (or sub-bands) used for data transmission and the frequency sub-band allocated for voice transmission.

Detailed Description Text (46):
Although the frequency sub-bands in the preferred embodiments may be varied, the particular choices illustrated in FIGS. 4-6 have the benefit of providing good voice transmission quality in all cases.

Detailed Description Text (47):
In addition, in those environments in which the analog transmission channel is a local telephone loop or a public switched telephone network circuit, call placement telephone signalling is required. In this instance the utilization of DTMF (dual tone multifrequency) and call progress signalling requires transmission of frequencies

within the range 350-1500 Hertz. In all of the cases indicated in FIGS. 4-6, the sub-band allocated for voice transmission is adequate to transmit without impairment these call placement signals. Thus, in all cases illustrated in FIGS. 4-6, no band switching during call placement is required, since the sub-band allocated for voice transmission in these instances is sufficient to transmit all telephone signals without impairment.

Detailed Description Text (48):
Data transmission in the sub-band above the voice is the only choice available in the parallel architecture embodiment illustrated in FIG. 9. This embodiment is particularly well suited to the case in which the analog channel is a long distance public switched telephone connection since frequencies below 300 Hertz are considerably attenuated on such channels and hence transmission in a sub-band below the voice is generally unsatisfactory.

Detailed Description Text (49):
On the other hand, in the embodiment illustrated in FIG. 2, in which the transmission channel is the local loop telephone circuit, there is no restriction on the lowest frequency available for transmission; and in this environment a data sub-band below voice, either alone or in conjunction with the data sub-band above voice, is well suited for data transmission.

Detailed Description Text (51):
The telephone set 19 illustrated diagrammatically at the right-hand side of the top of FIG. 7 and at the right-hand side of the bottom of FIG. 7 illustrates the respective on-hook (absence of speech) and off-hook (presence of speech) telephone conditions. When the telephone is on hook the entire telephone band is available and is shown in the top part of FIG. 7 as being used for the transmission of data. The lower part of FIG. 7 shows the off hook telephone for the simultaneous transmission of speech and data, and in this instance the data is restricted to a sub-band in the upper edge of the available channel.

Detailed Description Text (52):
While simultaneous transmission of data in an upper sub-band only is illustrated in the lower part of FIG. 7, the data could also be transmitted around the voice as illustrated in FIG. 4, or below the voice as illustrated in FIG. 6, in alternate preferred embodiments.

Detailed Description Text (56):
After allowing for filter transition regions the voice sub-band utilized by this example embodiment is between 350 Hz and 2450 Hz.

Detailed Description Text (57):
In evaluating the loop transmission characteristics the multiplexors 5 and 13 of FIG. 2 exchange information regarding these characteristics during a training phase. This information consists of the attenuation and background noise experienced by each of the carriers (which carriers are separated by 7.8125 Hz) transmitted within the data sub-bands. Based on this information, the multiplexors 5 and 13 allocate power to each carrier in such a fashion as to maximize data throughput. One consequence of this

process is the determination of the number of bits modulated onto each carrier, selected from the choices of 6, 4, 2, or 0 (i.e. unused).

Detailed Description Text (65):
The voice sub-band has been chosen so that all DTMF tones generated by the telephone, as well as signalling tones generated by the central office (dial tone, audible ringing tone, call waiting tone, confirmation tone, etc.) pass unimpaired through the multiplexors 5 and 13.

Detailed Description Text (70):
With the previously described voice and data sub-band settings, it is possible to transmit data through the telephone interface of the remote multiplexor using a 300 bps Bell 103 compatible modem. Should the user wish to employ a Bell 212 compatible modem it would be necessary to reduce the high data channel bandwidth. This can be accomplished either through commands issued to the remote multiplexor data port or through the addition of a switch to the remote multiplexor unit. A data port command or switch closure will then cause both multiplexors to redefine the lower edge of the upper data band and to redeploy the available power over the reduced band. U.S. application Ser. No. 06-736,200, now U.S. Pat. No. 4,679,227 is incorporated by reference.

CLAIMS:

1. A speech and data multiplexor for use over imparied and bandwidth restricted analog channels or digital representative of such channels, said multiplexor comprising,

allocating means for alllocating a speech sub-band in a central poriton of the available bandwidth for the transmission of speech,

selecting means for selecting at least one data sub-band, within the available bandwidth and on a border of the speech sub-band and in a portion of the available bandwidth separate and distinct from the portion of the bandwidth containing the speech sub-band, for the transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attenuation and/or group delay are common in the data sub-band, and

adaptive means for adapting the transmission of the data in the data sub-band to any impairments existing in the data sub-band and independent of any speech signals and/or channel impairments existing or occurring in the speech sub-band and effective to optimize the transmission of data in the data sub-band in the presence of said impairments in the data sub-band and within the limt of the power available or allocated for the transmission.

2. A speech and data multiplexor for use over impaired and bandwidth restricted analog channels or digital representatives of such channels, said multiplexor comprising,

allocating mean for allocating a speed sub-band in a central

portion of the available bandwidth for the transmission of speech,

selecting means for selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attentuation and/or group delay are common in the data sub-band,

adaptive means for adapting the transmission of the data in the data sub-band to any impairments existing in the data sub-band and effective to optimize the transmission of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission, and wherein the adaptive means include evaluating means for determination of the total available bandwith for data transmission, said evaluating means being operatively associated with the selecting means for said selection of the data sub-band.

3. A speech and data multiplexor for use over impaired and bandwidth restrited analog channels or digital representatives of such channels, said multiplexor comprising,

allocating means for allocating a speech sub-band in a central portion of the available bandwidth for the transmission of speech,

selecting means for selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attentuation and/or group delay are common in the data sub-band,

adaptive means for adapting the transmission of the data in the data sub-band to any impairments existing in the data sub-band and effective to optimize the transmission of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission, and wherein the adaptive means include multiple signal carriers uniformly spaced apart in small increments of frequency and wherein said adaptive means are effective both to select certain carriers for transmitting the data and also to select a particular modulation scheme for each selected carrier in order to optimize the overall rate of data transmission in the presence of any impairments existing in the data sub-band.

4. A speech and data multiplexor for use over impaired and bandwidth restricted analog channels or digital representatives of such channels, said multiplexor comprising,

allocating means for allocating a speech sub-band in a central portion of the available bandwidth for the transmission of speech,

selecting means for selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the

transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attentuation and/or group delay are common in the data sub-band,

adaptive means for adapting the transmission of the data in the data sub-band to any impairments existing in the data sub-band and effective to optimize the transmission of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission, and wherein the adaptive means include an ensemble modem which utilizes multiple signal carriers and a multimode modulation scheme for transmitting the data in the data sub-band.

6. The invention defined in claim 5 wherein the selection of the signal carriers adapts to induced power and telephone equipment noise in a low frequency data sub-band, provides carrier by carrier demodulation to permit compensation for frequency dependent attenuation in both low and high frequency data sub-bands, and provides a long symbol period modulation with an associated guardtime to make the transmission of data insensitive to frequency dependent group delay.

7. The invention defined in claim 4 wherein the ensemble modem includes testing means for testing the analog channel transmission characteristics during a training phase to determine the attenuation and the background noise experienced by each of the carriers transmitted within the data sub-band and wherein the modem then optimizes the transmission of data by deploying the available power over selected carrier frequencies and with a modulated number of bits on each selected carrier to obtain the most efficient use of the data sub-band for the transmission of data in view of the existing analog channel transmission characteristics.

8. The invention defined in claim 1 wherein the allocating means permit user selection of the boundaries of the speech sub-band.

9. The invention defined in claim 8 wherein the multiplexor includes a data port and the user selection of the speech sub-band to be utilized is accomplished by means of information transmitted through the data port.

10. The invention defined in claim 1 including transmitting means for transmitting speech in the speech sub-band.

11. The invention defined in claim 1 including transmitting means for transmitting data in the speech sub-band.

13. The invention defined in claim 1 wherein the selecting means select a single data sub-band above the speech sub-band.

14. The invention defined in claim 1 wherein the selecting means select a single data sub-band below the speech sub-band.

15. The invention defined in claim 1 wherein the selecting means select two data sub-bands, one data sub-band above the speech

sub-band and the other data sub-band below the speech sub-band.

16. A speech and data multiplexor for use over impaired and bandwidth restricted analog channels or digital representatives of such channels, said multiplexor comprising,

allocating means for allocating a speech sub-band in a central portion of the available bandwith for the transmission of speech,

selecting means for selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attentuation and/or group delay are common in the data sub-band,

adaptive means for adapting the transmisison of the data in the data sub-band to any impairments existing in the data sub-band and effective to optimize the transmission of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission, and wherein the selecting means select a single data sub-band above the speech sub-band and including detecting means for detecting the absence of speech transmission on the analog channel and effective in combination with the selecting means for selecting all of the available bandwith for the transmission of data in the absence of speech.

17. A speech and data multiplexor for use over impaired and bandwidth restricted analog channels or digital representatives of such channels, said multiplexor comprising,

allocating means for allocating a speech sub-band in a central portion of the available bandwith for the transmission of speech,

selecting means for selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attentuation and/or group delay are common in the data sub-band,

adaptive means for adapting the transmission of the data in the data sub-band to any impairments existing in the data sub-band and effective to optimize the transmission of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission, and wherein the selecting means select a single data sub-band below the speech sub-band and including detecting means for detecting the absence of speech transmission on the analog channel and effective in combination with the selecting means for selecting all of the available bandwidth for the transmission of data in the absence of speech.

18. A speech and data multiplexor for use over impaired and

bandwidth restricted analog channels or digital representatives of such channels, said multiplexor comprising,

allocating means or allocating a speech sub-band in a central portion for the available bandwidth for the transmission of speech,

selecting means for selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attentuation and/or group delay are common in the data sub-band,

adaptive means for adapting the transmission of the data in the data sub-band to any impairments existing in the data sub-band and effective to optmize the transmission of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission, and wherein the selecting means select two data sub-bands, one data sub-band above the speech sub-band and the other data sub-band below the speech sub-band and including detecting means for detecting the absence of speech transmission on the analog channel and effective in combination with the selecting means for selecting all of the available bandwidth for the transmission of data in the absence of speech.

19. The invention defined in claim 1 including separating means for separating the speech and data energies that are present in the speech sub-band and data sub-band.

21. The invention defined in claim 20 wherein the filter means include programmable, active filter means for selecting the edge frequencies of the sub-bands.

22. The invention defined in claim 21 wherein the multiplexor includes an ensemble modem operatively associated with the filter means for allowing the ensemble modem to select the edge frequencies of the sub-bands.

25. The invention defined in claim 24 wherein the speech sub-band is allocated in the frequency range of about 300 Hz to 2700 Hz.

26. A speech and data multiplexor for use over impaired and bandwidth restricted analog channels or digital representatives of such channels, said multiplexor comprising,

allocating means for allocating a speech sub-band in a central portion of the available bandwidth for the transmission of speech,

selecting means for selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attenuation and/or group delay are

common in the data sub-band,

adaptive means for adapting the transmission of the data in the data sub-band to any impairments existing in the data sub-band and effective to optimize the transmisison of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission, and wherein the analog channel is a telephone channel having a restricted bandwith in the general range of 0-4000 Hz and wherein the edges of the speech sub-band are allocated to permit all dual tone multi-frequency (DTMF) and call progress signalling without the need to switch the speech or data sub-bands.

28. The invention defined in claim 24 wherein the multiplexor has three ports--a channel port, a speech port and a data port--and filtering means for separating speech and data energies that are present in the sub-bands and wherein the filter means are active programmable filter means operatively associated with the data port for permitting user selection of the edges of the sub-bands to be employed by the multiplexor.

31. A speech and data multiplexor for use over impaired and bandwidth restricted analog channels or digital representations of such channels, said multiplexor comprising,

allocating means for allocating a speech sub-band in a cental portion of the available bandwidth for the transmission of speech,

selecting means for selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attenuation and/or group delay are common in the data sub-band,

adaptive means for adapting the transmission of the data in the data sub-band to any impairments existing in the data sub-band and effective to optimize the transmission of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission,

said adaptive means including evaluating means for determination of the total available bandwidth for data transmission, said evaluating means being operatively associated with the selecting means for said selecting of the data sub-band,

said adaptive means including an ensemble modem which utilizes multiple signal carriers and a multimode modulation scheme for transmitting the data in the data sub-band,

filter means for separating the speech and data energies that are present in the speech sub-band and data sub-band, said filter means including programmable, active filter means for selecting the edge frequencies of the sub-bands, and wherein the ensemble modem is operatively associated with the filter means for allowing the ensemble modem to select the edge frequencies of the sub-bands, and

wherein the analog channel is a telephone channel having a restricted bandwidth in the general range of 0-4000 Hz and the edges of the speech sub-band are allocated to permit all dual tone multi-frequency (DTMF) signalling without the need to switch the speech or data sub-bands.

32. A speech and data multiplexor for use over impaired and bandwidth restricted analog channels or digital representatives of such channels, said multiplexor comprising,

allocating means for allocating a speech sub-band in a central portion of the available bandwidth for the transmission of speech,

selecting means for selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attentuation and/or group delay are common in the data sub-band,

adaptive means for adapting the transmission of the data in the data sub-band to any impairments existing in the data sub-band and effective to optimize the transmission of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission, and including user input means for inputting a requested speech quality signal, a requested data rate signal or a value indicating the relative user weighting of speech quality and data rate and wherein said allocating means and selecting means determine the edge frequencies of the speech and data sub-bands in response to the signal which is input by said user input means.

33. A speech and data multiplexor for use over impaired and bandwidth restricted analog channels or digital representatives of such channels, said multiplexor comprising,

allocating means for allocating a speech sub-band in the middle part of the available bandwidth for the transmission of speech,

selecting means for selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the transmission of data,

said analog channel being an analog channel of the kind in which impairments of induced noise, attenuation and/or group delay are common in the data sub-band,

elvaluating means for determination of the total available bandwidth for data transmission, said evaluating means being operatively associated with the selecting means for said selecting of the data sub-band, and

multicarrier data transmitting means for transmitting data in the data sub-band on a plurality of carriers.

34. A method of multiplexing speech and data over impaired and bandwidth restricted analog channels or digital representations of such channels, said method comprising,

allocating a speech sub-band in the middle part of the available bandwidth for the transmission of speech,

selecting at least one data sub-band, within the available bandwidth and on a border of the speech sub-band and in a portion of the available bandwidth separate and distinct from the portion of the bandwidth containing the speech sub-band, for the transmission of data, and

adapting the transmission of the data in the data sub-band to the impairments of induced noise and attenuation that may be present in the data sub-band and independently of any speech signals and/or channel impairments existing or occurring in the speech sub-band to optimize the transmission of data in the data sub-band in the presence of said impcirments in the data sub-band and within the limit of the power available or allocated for the transmission.

35. The invention defined in claim 34 wherein the adapting step includes evaluating the total available bandwidth for selecting said data sub-band.

36. A method of multiplexing speech and data over impaired and bandwidth restricted analog channels or digital representations of such channels, said method comprising,

allocating a speech sub-band in the middle part of the available bandwidth for the transmission of speech,

selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the the transmission of data,

adapting the transmission of the data in the data sub-band to be impairments of induced noise and attenuation that may be present in the data sub-band to optimize the transmission of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission, and including utilizing multiple signal carriers uniformly spaced apart in small increments of frequency for transmitting data in the data sub-band and including selecting certain carriers for the transmission of data and also selecting a particular modulation scheme for each selected carrier (to determine the number of data bits modulated on each carrier) for said optimization of the overall rate of data transmission in the presence of any impairments existing in the data sub-band.

37. A method of multiplexing speech and data over impaired and bandwidth restricted analog channels or digital representations of such channels, said method comprising,

allocating a speech sub-band in the middle part of the available bandwidth for the transmission of speech,

selecting at least one data sub-band within the available bandwidth and on a border of the speech sub-band for the the transmission of data,

adapting the transmission of the data in the data sub-band to be impairments of induced noise and attenuation that may be present in the data sub-band to optimize the transmission of data in the data sub-band in the presence of said impairments and within the limit of the power available or allocated for the transmission, and including determining the edge frequencies of the speech and data sub-bands in response to a user requested speech quality, a user requested data rate or a value indicating the realtive user weighting of speech quality and data rate.